L3: Entry 2 of 29 File: TDBD Dec 1, 2001

TDB-ACC-NO: NNRD452114

DISCLOSURE TITLE: Dialing Proxy Service Center

PUBLICATION-DATA:
IBM technical Disclosure Bulletin, December 2001, UK

ISSUE NUMBER: 452
PAGE NUMBER: 2110

PUBLICATION-DATE: December 1, 2001 (20011201)

CROSS REFERENCE: 0374-4353-0-452-2110

DISCLOSURE TEXT:

Disclosed is a proxy service for dialing a telephone number. Users of the service register their telephone number lists with the proxy service center. The users can make telephone calls merely by dialing the proxy service center. The users do not have to carry their telephone number lists. The service would be very useful for the blind, the elderly, and people who have heart pacemakers, because some of them can not use cellular phones (which include telephone number lists). The service is explained based on the user scenario below (See Fig. 1). 1) Users register their personal telephone number lists (e.g., telephone numbers of hospitals, grandchildren, etc.) to the proxy service center in advance.

The following should be registered for each telephone number: Full name Nickname (optional) (e.g., "Bill" for William, "My son", etc.) Telephone number Full name or nickname pronunciation pronounced by the users themselves to improve voice recognition accuracy (optional) 2) The user makes a telephone call to the Dialing Proxy Service Center. When users call the center using the telephone in their own homes, the center can identify who is calling based on the registered users' home telephone numbers. The center then finds the personal telephone number list registered by the user.

When users make calls to the center using any telephone besides their own home phones, they dial their own user numbers. The center indexes the personal telephone number lists by the user numbers. 3) The users say the full name or nickname of the person they want to make a call to. The Voice Recognition Server or the operator will recognize who the user wants to make the call to. The users can use the full names or nicknames for the calls. Basically, the voice recognition server recognizes either the full name or the nickname. Users can register their own personal pronunciations of the full names and the nicknames in order to improve the accuracy of the voice recognition.

If the line has a lot of noise and it is difficult for the Voice Recognition Server to recognize the full name or the nickname, a human operator instead of the Voice Recognition Server may be able to recognize it. 4) The Dialing Server dials the correct phone number, and it transfers the phone call to the user. The Dialing Server dials the destination phone number according to the Personal Telephone Number List, and transfers the telephone call to the user.

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☐ 1. Document ID: NN941027

L4: Entry 1 of 4 File: TDBD Oct 1, 1994

TDB-ACC-NO: NN941027

DISCLOSURE TITLE: Communications Enhancements Made Possible by Caller-ID

Full | Title | Citation | Front | Review | Classification | Date | Reference | Sequences | Attachments | Claims | KWIC | Draw Desc

☐ 2. Document ID: NN9401391

L4: Entry 2 of 4 File: TDBD Jan 1, 1994

TDB-ACC-NO: NN9401391

DISCLOSURE TITLE: Method for Automatic Directory Update upon Encounter of Specific Callee Messages

Full | Title | Citation | Front | Review | Classification | Date | Reference | Sequences | Attachments | Claims | KWIC | Draw Desc

☐ 3. Document ID: NN9303309

L4: Entry 3 of 4 File: TDBD Mar 1, 1993

TDB-ACC-NO: NN9303309

DISCLOSURE TITLE: Accounting System for Rolm 244 PC Adapter using Caller Voice Recognition

Full | Title | Citation | Front | Review | Classification | Date | Reference | Sequences | Attachments | Claims | KWIC | Draw Desc

☐ 4. Document ID: NN86044897

L4: Entry 4 of 4 File: TDBD Apr 1, 1986

TDB-ACC-NO: NN86044897

DISCLOSURE TITLE: Computerized Call Return Feature

Full | Title | Citation | Front | Review | Classification | Date | Reference | Sequences | Attachments | Claims | KWIC | Draw Desc | Clip Img

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voice adj3 recognition and callee	4

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Session Initiation Protocol (SIP)

SIP, the Session Initiation Protocol, is a signaling protocol for Internet conferencing, telephony, presence, events notification and instant messaging. SIP was developed within the IETF MMUSIC (Multiparty Multimedia Session Control) working group, with work proceeding since September 1999 in the IETF SIP working group.

A number of standardization organizations and groups are using or considering SIP:

- IETF PINT working group
- 3GPP for third-generation wireless networks
- Softswitch Consortium
- IMTC and ETSI Tiphon are working on SIP-H.323 interworking
- PacketCable DCS (distributed call signaling) specification

[News] [Overview] [Where is SIP being discussed?] [Search] [What SIP extensions are being planned?] [Drafts] [SIP grammar] [Implementations] [SIP service providers] [Vendors] [Mailing list] [Public SIP Servers] [Papers] [Talks] [Related Drafts and Documents] [Draft History] [Frequently Asked Questions (FAQ)] [Port Assignments and DNS] [Compact headers] [SIP-related events] [SIP interoperability testing events] [Press Coverage] [Emergency Calling] [Non-Internet-related SIP sightings] [Status and Schedule] [Internships and Jobs] [Basic telephony features] [Other SIP sites]

Last updated 07/07/2004 16:09:02 by Henning Schulzrinne

SIP Vs. H.323 - A Comparison

As a manufacturer of telephone test equipment, we have to evaluate the potential market for both H.323 and SIP. H.323 is the more mature of the two, but problems may arise due to lack of flexibility. SIP is currently less defined, but has greater scalability which could ease internet application integration. Which protocol will win out in the end? It is still too early to tell, but our unbiassed analysis will help you decide which protocol best suites your application.

	H.323	SIP
Architecture	H.323 covers almost every service, such as capability exchange, conference control, basic signaling, QoS, registration, service discovery, and so on.	SIP is modular because it covers basic call signaling, user location, and registration. Other features are in other separate orthogonal protocols.
Components	Terminal/Gateway	UA
	Gatekeeper	Servers
Protocols	RAS/Q.931	SIP
	H.245	SDP
Call control Functionality		
Call Transfer	Yes	Yes
Call Forwarding	Yes	Yes
Call Holding	Yes	Yes
Call Parking/Pickup	Yes	Yes
Call Waiting	Yes	Yes
Message Waiting Indication	Yes	No
Name Identification	Yes	No
Call Completion on Busy Subscriber	Yes	Yes
Call Offer	Yes	No
Call Intrusion	Yes	No
	H.323 splits them across H.450, RAS, H.245 and Q.931	

Advanced Features		
Multicast Signaling	Yes, location requests (LRQ) and auto gatekeeper discovery (GRQ).	Yes, e.g., through group INVITEs.
Third-party Call Control	Yes, through third-party pause and re-routing which is defined within H.323. More sophisticated control is defined by the related H.450.x series of standards.	Yes, through SIP as described in separate Internet Drafts.
Conference	Yes	Yes
Click for Dial	Yes	Yes
Scalability		
Large Number of Domains	The initial intent of H.323 was for the support of LANs, so it was not inherently designed for wide area addressing. The concept of a zone was added to accommodate wide area addressing. Procedures are defined for "user location" across zones for email names. Annex G defines communication between administrative domains, describing methods to allow for address resolution, access authorization and usage reporting between administrative domains. In multi-domain searches, there is no easy way to perform loop detection. Performing the loop detection can be done (using the PathValue field), but introduces other issues related to scalability.	SIP inherently supports wide area addressing. When multiple servers are involved in setting up a call, SIP uses a loop detection algorithm similar to the one used in BGP, which can be done in a stateless manner, thus avoiding scalability issues. The SIP Registrar and redirect servers were designed to support user location.
Large Number of Calls	H.323 call control can be implemented in a stateless manner. A gateway can use messages defined in H.225 to assist the gatekeeper in performing load balancing across gateways.	Call control can be implemented in a call stateless manner. SIP supports n to n scaling between UAs and servers. SIP takes less CPU cycles to generate signaling messages; therefore a server could theoretically handle more transactions. SIP has specified a method of load balancing based upon the DNS SRV record translation mechanisms.
Connection State	Stateful or Stateless.	Stateful or Stateless. A SIP call is independent of the existence of a transport-layer connection, but instead signals call termination explicitly.
Internationalization	Yes, H.323 uses Unicode (BMPString within ASN.1) for some textual information (h323-id), but generally has few textual parameters.	Yes, SIP uses Unicode (ISO 10646-1), encoded as UTF-8, for all text strings, affording full character set neutrality for names, messages and parameters. SIP provides for the

		indication of languages and language preferences.
Security	Defines security mechanisms and negotiation facilities via H.235, can also use SSL for transport-layer security.	SIP supports caller and callee authentication via HTTP mechanisms. Cryptographically secure authentication and encryption is supported hop-by-hop via SSL/TSL, but SIP could use any transport-layer or HTTP-like security mechanism, such as SSH or S-HTTP. Keys for media encryption are conveyed using SDP. SSL supports symmetric and asymmetric authentication. SIP also defines end-to-end authentication and encryption using either PGP or S/MIME.
Interoperability among Versions	The fully backward compatibility in H.323 enables all implementations based on different H.323 versions to be seamlessly integrated.	In SIP, a newer version may discard some old features that are not expected to be implemented any more. This approach saves code size and reduces protocol complexity, but loses some compatibility between different versions.
Implementation Interoperability	H.323 provides an implementers' guide, which clarifies the standard and helps towards interoperability among different implementations.	SIP thus far has not provided an implementation agreement.
Billing	Even with H.323's direct call model, the ability to successfully bill for the call is not lost because the endpoint reports to the gatekeeper the beginning and end time of the call via the RAS protocol.	If the SIP proxy wants to collect billing information, it has no choice but to stay in the call signaling path for the entire duration of the call so that it can detect when the call completes. Even then, the statistics are skewed because the call signaling may have been delayed.
Codecs	H.323 supports any codec, standardized or proprietary, not just ITU-T codecs. There have been codepoints for MPEG and GSM, which are not ITU-T codecs, in H.323 for a long time; many vendors support proprietary codecs through ASN.1 NonStandardParameters, which is equivalent to SIP's "privately-named codec by mutual agreement"; and any codec can be signaled via the GenericCapability feature that was added in H.323v3. Payload types can be specified statically or dynamically.	SIP supports any IANA-registered codec (as a legacy feature) or other codec whose name is mutually agreed upon. Payload types can be specified statically or dynamically.
Call Forking	H.323 gatekeeper can control the call signaling and may fork the call to any number of devices simultaneously.	SIP proxies can control the call signaling and may fork the call to any number of devices simultaneously.

Transport protocol	Reliable or unreliable, e.g., TCP or UDP. Most H.323 entities use a reliable transport for signaling.	Reliable or unreliable, e.g., TCP or UDP. Most SIP entities use an unreliable transport for signaling.
Message Encoding	H.323 encodes messages in a compact binary format that is suitable for narrowband and broadband connections.	SIP messages are encoded in ASCII text format, suitable for humans to read.
Addressing	Flexible addressing mechanisms, including URLs and E.164 numbers.	SIP only understands URL-style addresses.
PSTN Interworking	H.323 borrows from traditional PSTN protocols, e.g., Q.931, and is therefore well suited for PSTN integration. However, H.323 does not employ the PSTN's circuit-switched technology--like SIP, H.323 is completely packet-switched. How Media Gateway Controllers fit into the overall H.323 architecture is well-defined within the standard.	SIP has no commonality with the PSTN and such signaling must be "shoe-horned" into SIP. SIP has no architecture that describes the decomposition of the gateway into the Media Gateway Controller and the Media Gateways.
Loop Detection	Yes, routing gatekeepers can detect loops by looking at the CallIdentifier and destinationAddress fields in call-processing messages. If the combination of these matches an existing call, it is a loop.	Yes, the SIP message Via header facilitates this. However, there has been talk about deprecating Via as a means of loop detection due to its complexity. Instead, the Max-Forwards header seems to be the prefered method of limiting hops and therefore loops.
Minimum Ports for VoIP Call	5 (Call signaling, 2 RTP, and 2 RTCP.)	5 (Call signaling, 2 RTP, and 2 RTCP.)
Video and Data Conferencing	H.323 fully supports video and data conferencing. Procedures are in place to provide control for the conference as well as lip synchronization of audio and video streams.	SIP has limited support for video and no support for data conferencing protocols like T.120. SIP has no protocol to control the conference and there is no mechanism within SIP for lip synchronization.
Microtronix Test System Available	Yes	Yes

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